REMARKS/ARGUMENT

In the Office Action, the Abstract of Disclosure is objected to. In response, Applicant has amended the Abstract to incorporate the language suggested by the Examiner. Approval of the amended Abstract and withdrawal of this objection is respectfully requested.

Claim 1 stands rejected under 35 U.S.C. §102(b) as being anticipated by U.S. Patent 5,559,891 to Kuusama (Kuusama '891) or, alternatively, by U.S. Patent 6,141,415 to Rao (Rao '415). The Examiner's rejection on this ground is respectfully traversed.

Among the limitations of independent claim which are neither taught nor suggested in Kuusama are a loudspeaker unit comprising:

> a processor for *comparing* in real time an output signal from the microphone with an output signal from a sound source with reference to a frequency characteristic and an echo characteristic, or a reverberation characteristic, and *correcting* a signal from the sound source using the *difference in output signal between the microphone and the sound source* by reference to the frequency characteristic and the echo characteristic or the reverberation characteristic". (emphasis added)

As a result of comparison of the output microphone and sound source signals and use of the difference between these output signals, desirable acoustic effects, such as an echo or reverberations, can be controllably modified every time the claimed loudspeaker unit is placed in a new environment. Therefore, and as opposed to systems that use a stored reference value, the inventive loudspeaker unit is an adaptive system that controls the above-mentioned acoustic effects unique to a given environment. Accordingly, a portion of the output signal of the sound source is stored and then compared to a regenerative feedback signal corresponding to the stored portion. As a result of comparison, the output signal of the sound source can be modified by either canceling out a difference to eliminate acoustic effects, or an echo and further reverberations can be generated in accordance with the user's requirements.

In contrast to the foregoing, Kuusama teaches canceling room resonances. After that, desired room effects are generated based on undisclosed algorithms. The structure described by Kuusama has no feedback to compare original and reproduced signals, as shown in FIGS. 5 and 6 and disclosed in column 4, lines 1-10 and 25-35.

Furthermore, no suggestion is provided in Kuusama that such the feedback can be used in both a structure shown in FIG. 7 and a relevant portion of the Kuusama's disclosure. In fact, except for addition of a sound source, as stated in column 4, lines 34-35, "the rest of the system is similar to that shown in FIG. 5 when both input signals are received through the microphone..." Therefore, Kuusama neither suggests nor teaches comparing output signals from the microphone and sound source to correct a signal from the sound source in response to determination of the difference in output signals. Accordingly, Kuusama does not clearly teach a processor which compares the original and reproduced signals, as claimed, nor does this reference suggest this element. Accordingly, withdrawal of this rejection is respectfully requested.

Claim 2 has been rejected under 35 U.S.C. §103 as being unpatentable over Kuusama '891. This rejection should be withdrawn.

As stated above, Kuusama neither teaches nor suggests the processor comparing the original and output signals for modifying the original signal based on a processed difference, as recited in claim 1 of invention. Claim 2 depending directly from claim 1 contains additional limitations that are neither taught nor suggested by Kuusama. Particularly, the processor includes two separate A/D converters one for an original signal and one for feedback signal. Therefore, the Applicant's structure processes a limited data, which is adequate for the purposes of the present invention.

Kuusama neither teaches nor suggests desperately digitizing two signals. As a result, Kuusama has to process a substantial amount of data, and, thus, does not even recognize one of the problems solved by the structure, as recited in claim 2.

The Examiner has contended that it is within the ability of one with ordinary skill in the art to use a second A/D converter in the Kuusama's structure. The Examiner's position on this ground is traversed.

Applicant submits that the Examiner's contention is an unfounded, undocumented conclusion since there is no direction, incentive, motivation or suggestion in Kuusama that the second A/D converter can or ought to be incorporated in this reference. There is no evidence submitted by the Examiner to support his conclusion.

Accordingly, the rejection under 35 U.S.C, 103 is respectfully requested to be withdrawn.

The specification has been revised to more sharply disclose the inventive scope of this invention. Care has been taken to avoid any introduction of new matter.

In view of the foregoing, favorable consideration of the newly submitted abstract, amended specification and allowance of the original claims is earnestly solicited.

I hereby certify that this correspondence is being deposited with the United States Postal Service with sufficient postage as First Class Mail in an envelope addressed to: Commissioner of Patents and Trademarks, Washington, D.C. 20231, on August 23, 2001

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Respectfully submitted,

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APPENDIX B

VERSION WITH MARKINGS TO SHOW CHANGES MADE
37 C.F.R. § 1.121(b)(iii) AND (c)(ii)

SPECIFICATION:

Paragraph at page 5, line 9, to line 10:

Next, the [motion] operation of the circuit of Fig. 2 will be described with reference to the drawing.

Paragraph at page 5, line 11, to page 6, line 3:

Sound source signal 7 of sound source 2 to be desirably regenerated is inputted to processor module 3 before it is inputted to amplifier 4. Processor module 3 compares feedback signal 9 inputted from microphone 6 with sound source signal 7. Processor module 3 [operates] applies correction data [so that] to the sound source signal to produce a correction signal and which applied to amplifier 4 and will cause loudspeaker 5 to generate a sound (which in turn causes microphone 6 to generate a feedback signal 9) [may come most] which is as close as possible to the original sound source signal 7. [for the sake of obtaining] As a result, the system will obtain a reasonable sound intensity characteristic or the desirable effect of echo suppression. [and by applying thus obtained result to inputted sound source signal 7, produces correction signal 8 to send to amplifier 4. Amplifier 4 amplifies correction signal 8 to produce the sound from loudspeaker 5.] Since [this] the sound generated by loudspeaker 5 has been corrected [at] in real time with reference to the frequency characteristic or the reverberation characteristic [affected by the property of the installation place of] characteristic of the room in which the loudspeaker [unit] 5 is located, correction signal 8 approaches sound source signal 7.

Paragraph at page 6, line 4, to line 19:

Next, a concrete embodiment of the present invention will be described in detail referring to the drawings. With reference to Fig. 3, sound source 2 is [a sounder] any sound source such as a radio tuner, a compact disk or a sound chip of a personal computer. Processor module 3 comprises 16 bit A/D converter 31, 16 bit A/D converter 32, digital signal processor 35, 16 bit

D/A converter 33, and memory 34. Amplifier 4 is on operational amplifier. It drives loudspeaker 5 to 57 mm in diameter with impedance of 8 Ω. Microphone 6 is composed of an electret condenser microphone of 9.5 mm in diameter with a flat frequency characteristic and a microphone amplifier. A cable which transmits feedback signal 9 outputted from microphone 6 is selected from a group of the noise-resistant shielding wire.

Paragraph at page 6, line 20, to line 22:

Next, the [motion] <u>operation</u> of the embodiment of the present invention will be described in detail with reference to Fig. 3.

Paragraph at page 6, line 23, to page 8, line 2:

Signal 7 from sound source 2 is converted to a digital signal by A/D converter 31 of processor module 3 and stored in memory 34. The data of all signals A/D converted within a fixed time stipulated for the reverberation and the echo are stored as the data of sound source 2 in memory 34. On the other hand, a signal processed as a regenerative signal by digital signal processor 35 of processor module 3 is further converted to an analog signal by means of D/A converter 33, [and after] amplified by amplifier 4, [it is sent forth from] <u>and applied to</u> loudspeaker 5 [as a] <u>which generates a corresponding</u> sound. Microphone 6 picks up this sound<u>,</u> <u>and converts it into a</u> [then the sound is converted as] feedback signal 9 <u>which is converted into</u> [to] a digital signal by A/D converter 32 and inputted to digital signal processor 35. Successive comparison analysis part 37 of digital signal processor 35 compares the data of sound source 2 stored in memory 34 with digital data from successive A/D converter 32, analyzes the intensity of the reverberation and the echo, corrects the conversion data stored in memory 34 and [gets] <u>obtains</u> a correction parameter. Regenerative signal processing part 36 adds the correction parameter to the conversion data of sound source 2 and processes the digital data to regenerate <u>it</u> as a regenerative signal. The difference between the data of sound source 2 and the data of feedback signal 9 is obtained as the correction parameter in serial data and the parameter is processed by adding feedback signal 9 of an opposite phase, if necessary, to obtain a fixed number

of 0. The processed signal is converted to an analog signal by D/A converter 33, amplified by amplifier 4 and then sent forth from loudspeaker 5 as the sound.

CLAIMS:

3. (Amended) A loudspeaker unit adapted to the environment according to Claim [2] 1 wherein said successive comparison analysis part performs processing by adding antiphase feedback data to said voice data so that the difference between said voice data obtained as the serial data and said feedback data becomes a fixed value or 0.

5. (Amended) A loudspeaker unit adapted to the environment according to Claim [2] 1 wherein, the frequency comparison of the characteristic and the comparison of the characteristic of the echo or the reverberation each including the delay time are learned by arithmetic and a signal to be sent to the loudspeaker is corrected according to the learned result.

7. (Amended) A loudspeaker unit adapted to the environment according to Claim [2] 1 wherein, the frequency comparison of the characteristic and the comparison of the characteristic of the echo or the reverberation each including the delay time are intermittently performed and a signal to be sent to the loudspeaker is corrected according to the comparison result.

ABSTRACT

[The present invention provides a] A loudspeaker unit which requires no particular procedure for correction of the acoustic characteristic even if the installation environment of the loudspeaker unit changes, and which can correct, in addition to the frequency characteristic, a sound lag and a phase shift ascribable to the reverberation and an echo of a sound. The loudspeaker unit picks up a sound regenerating from the loudspeaker with a microphone, and compares [at] in real time a sound from a sound source with a regenerative sound, referring to a difference therebetween, with reference to the characteristic at an optional frequency and the characteristic of the reverberation or the echo each including the delay time, respectively, and corrects the signal to be sent to the loudspeaker by the result of arithmetic. [Further, correction of signals can be learned through arithmetic, and also correction of signals can be made by utilizing intermittent arithmetic.]